CLAIMS

What is claimed is:

1. A method for interfacing a Public Switched Telephone Network (PSTN) with a Voice over IP (VoIP) enabled access network comprising the steps of:

(a) receiving incoming call signaling from a PSTN, wherein the incoming call signaling is in a digital trunk format;

(b) converting the call signaling to a packet-based VoIP call signaling message stream; and

(c) transmitting the packet based VoIP call signaling stream to a VoIP receiving device.

2. The method of claim 1, further comprising the steps of.

(d) receiving the packet-based VoIP call signaling at a VoIP receiving device; and

(e) generating signaling compatible with a residential PSTN phone device.

3. The method of claim 1, wherein the incoming call is in a GR-303 format.

4. The method of claim 1, wherein the incoming call signaling is in an ETSI V5 interface format.

5. A method for transporting ring control signals between a PSTN and a VoIP enabled access network so as to minimize delay and maintain caller ID timing, the method comprising the steps of:

(a) receiving robbed bit signaling from a PSTN, wherein the robbed bit signaling contains the ring control signals

(b) converting the robbed bit signaling to specialized packets in a VoIP signaling stream without parsing the robbed bit signaling to produce a high level ring command

(c) transmitting the specialized packets over a VoIP enabled access network.

6. The method of claim 5, further comprising the steps of:

(d) receiving the specialized packets at a VoIP enabled device; and

(e) converting the specialized packets to a series of PSTN end user device compatible signals.

7. The method of claim 5, wherein the timing relationship between the robbed bit signaling and the bearer channel traffic is sustained.

8. A system for interfacing a Public Switched Telephone Network (PSTN) with a Voice over IP (VoIP) enabled access network, comprising:

a local digital switch (LDS) application for a receiving incoming call signaling from a PSTN, wherein the incoming call signaling is in a digital trunk format;

a converter for converting the call signaling to a packet-based VoIP call signaling message stream; and

a VoIP application for transmitting the packet based VoIP call signaling stream to a VoIP receiving device.

9. The system of claim 8, whereby the VoIP application receives the packet-based VoIP call signaling and the LDS application generates signaling compatible with a residential PSTN phone device.

10. The system of claim 8, wherein the incoming call is in a GR-303 format.

11. The method of claim 8, wherein the incoming call signaling is in an ETSI V5 interface format.

12. The system of claim 8, whereby the converter further includes a signaling converter for processing control signals and a voice converter for processing voice signals.

13. A system for transporting ring control signals between a PSTN and a VoIP enabled access network so as to minimize delay and maintain caller ID timing, comprising:

a local digital switch (LDS) application for receiving robbed bit signaling from PSTN, wherein the robbed bit signaling contains the ring control signals;

a converter for converting the robbed bit signaling to specialized packets in a VoIP signaling stream without parsing the robbed bit signaling to produce a high level ring command; and

a VoIP application for transmitting the specialized packets over said VoIP enabled access network.

14. The system of claim 13, whereby the VoIP application receives the specialized packets and the converter converts the specialized packets to a series of PTSN end user device compatible signals.

15. The system of claim 13, whereby the converter further includes a signaling converter for processing control signals and a voice converter for processing voice signals.

16. An internet protocol digital terminal for interfacing a Public Switched telephone Network (PSTN) with a Voice over IP (VoIP) enabled network, comprising:

a first interface for receiving TDMA communications comprising voice and signaling information from said PSTN and providing the voice and signaling information to a converter; and

for receiving voice and signaling information from said converter and for transmitting TDMA communications to said PSTN;

a second interface for receiving VoIP communications comprising voice and signaling information from said VoIP enabled network and providing voice and signaling information to said converter; and

for receiving voice and signaling information from said converter and transmitting said voice and signaling information to said VoIP enabled network;

whereby said converter converts TDMA-based voice and signaling information to VoIP-based voice and signaling information and converts VoIP -based voice and signaling information to TDMA-based voice and signaling information.